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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/678,222	10/06/2003	Mark Walter Chamberlain	HAR65 045	1579

77617 7590 06/25/2008

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EXAMINER	
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ART UNIT	PAPER NUMBER
2626	

MAIL DATE	DELIVERY MODE
06/25/2008	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/678,222	CHAMBERLAIN, MARK WALTER	
	Examiner	**Art Unit**	
	JOSIAH HERNANDEZ	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *06 October 2003*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-6,8-29 and 31-44* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-6,8-29 and 31-44* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *06 October 2003* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. Applicant's arguments filed 05/20/2008 have been fully considered. Applicant's request for reconsideration of the finality of the rejection of the last Office action is persuasive and, therefore, the finality of that action is withdrawn.

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1-10, 19, and 22, are rejected under 35 U.S.C. 103(a) as being unpatentable over Johnson (US 6,415,253) in view of Adlersberg et al. (US 5,012,519) and in further view of Ashley (US 6,453,291).

As to claim 1, Johnson discloses a method of reducing a noise component of an input speech signal (noise suppression device Abstract lines 1-3) comprised of

signal frames on a channel (dividing the received signal into data frames Abstract lines 3-5) comprising the steps of: applying a windowed Fourier transformation to said signal frames (applying fast Fourier transform to the appended data frames, abstract lines 7-9); approximating signal magnitudes of said signal frames (producing sets of magnitude components of the frequency spectrum for each of the frames, column 7 lines 20-24); computing Signal-to-Noise Ratio magnitudes of said signal frames (in spectral subtraction a signal-to-noise ratio is calculated by considering the magnitudes of the speech and noise signal, column 1lines 55-60); detecting voice activity in said channel (The average noise energy values are used to compare to the current noise energy value and determine, based on the En, the type of noise that exist in each frame in order to suppresses noise in a spectral component, column 8 lines 53-58, this technique is used to distinguish high level SNR signals that depict noise, column 8 lines 65-67, and further uses another technique to distinguish low level SNR signals that may contain such signals as silence, column 9 lines 1-5, Johnson teaches specifically that in the spectral subtraction techniques, the gain factors are adjusted by SNR estimates and the SNR estimates are determined by the speech energy in each frequency component and the current background noise energy estimate in each frequency component, column 2 lines 51-55, finally, Johnson teaches that the VAD computes energy ratios, column 9 lines 19-21, that are compared to threshold values in order to determine if there is noise in the signal, column 9 lines26-30, the ratios are determined by equation 5 and 6, of which are dependent on the previous ratios and the calculation

updates the ratio based on a new frame); detecting noise activity in said channel (noise activity, which is the noise signal, is detected from the input channel for the use of signal-to-noise ratio, column 1 lines 57 and 58; column 2 line 61); estimating gain in said signal frames, producing gain multiplicative factors based on the noise spectral estimate and frequency spectrum components (the frequency components are received from the windowed frame of signals, see column 3 lines 50 -57, the system uses a noise suppression spectral modifier, of which, produces gain multiplicative factors based on the noise spectral estimate and the frequency spectrum components, column 3 lines 55-58); applying said spectral gain function to the components of said windowed Fourier transformation (applying the gain factor to the frequency components, column 3 lines 57-60); and, applying an inverse Fourier transform to said signal frames thereby reconstructing a noise reduced output signal frame (once the noise has been reduced from the signal an inverse Fast Fourier Transform, column 1 lines 62-64, is used to convert the frequency components of the frames into time domain and reconstructing the noise reduced signal, column 3 lines 59-61.

Johnson does not specifically disclose applying an estimated noise history to signal frame.

Adlersberg teaches a noise reduction system (see title), of which applies an estimated noise history to signal frames to compute a spectral gain function, the gain function that is calculated depends on the adaptive noise estimates, which include averages of historical values (column 2 lines 50-58).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method of Johnson with historic SNR values for gain function calculation as disclosed by Adlersberg. Doing so would have given a more accurate noise estimator for the gains in each frame that change according to the SNR values and would have allowed for speech smoothing from frame to frame (column 2 lines 65-67).

Johnson or Adlersberg do not disclose specifically detecting voice activity in said channel as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ration thresholds.

Ashley teaches a system that overcomes the problem of fluctuation of non-stationary noise in speech signals (abstract) by using a Voice Activity Detection system for detecting voice in a signal (column 2 lines 31-33) the system then can detect voice activity by tracking the SNR of incoming signals and determines VAD SNR threshold of which is then biased using average variation in the incoming SNR values. This is done by estimating a SNR of an input signal (column 2 lines 34-36) and a threshold value is generated by the system based on the estimated SNR and biasing the VAD threshold based on a variation of the estimated SNR (column lines 36-38). Next, the threshold values are used to compare to the incoming calculated SNR values (column 8 lines 29-35). Finally, the averaged long-term power spectral estimates are performed (column 6 lines 62-64) in order to calculate the spectral deviation between the power spectrum and the average long-term power spectral estimate (column 6 lines 55 and 56). The deviation estimates are calculated from

the total energy estimates that are derived from the channel energy estimate calculations (column 6 lines 25-27) of which are used to calculate the signal to noise ratio from the channel energy (column 5 lines 55-60).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method of Johnson and Adlersberg with the method of detecting voice activity in said channel as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ratio thresholds as taught by Ashley. Using thresholds of average SNR ratios to compare to the incoming SNR from the signal allows for the threshold value to vary according to the specific noise environment that change and a consistent threshold value would not function correctly in an environment such as a car that outputs non-stationary noise. In this environment the VAD proposed will avoid loss of channel capacity or signal for CDMA variable rate coding as well as loss of signal for fixed-rate TDM with DTX (Ashley, column 1 lines 26-37) and a threshold value that is geared to decrease noise of non-stationary noise (Ashley, abstract) increases the ability to reduce noise in an environment such as in a car.

As to claim 2, Johnson does not specifically disclose using a database for stored historical SNR values.

Adlersberg teaches using current and previous noise estimates (see column 2 lines 58 &59), their storage implies using a database.

It would have been obvious to one having ordinary skill in the art at the time the invention was made that if more accurate calculations are made for SNR from historical values, these historical values would have to be stored in a list or in some organized fashion that constitute a database.

As to claim 3, Johnson discloses estimating noise values from signal frames, in the abstract it clearly states that an input signal is converted into frames and the signal is processed as frames from the beginning of the process until the end, (a SNR value is generated for the speech in each frequency component, the frequency components are received from the windowed frame of signals, column 3 lines 50 and 51 and column 1 lines 57 and 58).

Johnson does not specifically disclose applying an estimated noise history to signal frame.

Adlersberg teaches a noise reduction system (see abstract), of which applies an estimated noise history to signal frames to compute a spectral gain function (the gain function that is calculated depends on the prior and posterior signal-to-noise ratios, which are historical values, column 7 lines 45-50);

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method of Johnson with historic SNR values for gain function calculation as disclosed by Adlersberg. Doing

so would have given an accurate statistical estimator of how the gains in each frame should change according to the past SNR values.

As to claim 4, Johnson discloses using signal frames that are overlapped and added to previous signal frames (Johnson teaches overlapping adjacent frames, column 15 lines 33-36, and using the immediately previous frame, column 15 lines 41-46).

As to claim 5, Johnson discloses filtering said signal-to-noise ratio magnitude and signal magnitude prior to detecting voice activity in channel (Johnson teaches using a filter to process SNR magnitude values, which come from the signal and noise magnitude values. These values are then used to calculate a gain value in order to apply to the frequency spectrum once the voice activity has been detected, at this point the gain ratio would be applied to the voice activity, once the SNR values are detected the gain values are calculated in order to use on the components when undesirable speech is detected, which is done by the VAD, column 4 lines 1-7, therefore the SNR values are filtered before detecting voice activity, column 1 lines 52-60).

As to claim 6, Johnson discloses applying a windowed Fourier transform (a transform obtains frequency spectrum components from the windowed frame of signals, column 3 lines 49-52) on noise reduced output signal frame (once the

noise has been reduced from the signal an inverse transformer, of which is an inverse Fast Fourier Transform, column 1 lines 62-64, is used to convert the frequency components of the frames into time domain and reconstructing the noise reduced signal, column 3 lines 59-61).

As to claim 8, Johnson does not specifically disclose said noise component is Gaussian.

Adlersberg teaches reducing noise signal from a speech signal; the noise signal being a Gaussian signal (column 7 lines 25-28).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the Gaussian signal in Adlersberg. Gaussian noise signals are a very common noise signal and in order to effectively reduce all noise signals, Gaussian noise signals would also have to be included.

As to claim 9, Johnson discloses ramped noise (a non-stationary noise like a car passing at a distance; Johnson teaches non-stationary noise that can come from a passing car, column 3 lines 16-19 and column 5 lines 15-22).

As to claim 10, Johnson discloses non-stationary noise (a non-stationary noise like a car passing at a distance; Johnson teaches non-stationary noise that can come from a passing car, column 3 lines 16-19 and column 5 lines 15-22).

As to claim 19, Johnson discloses noise reduced output signal frame is overlapped and added to previous noise reduced output signals frame (Johnson teaches overlapping adjacent frames, column 15 lines 33-36) and using the immediately previous frame (column 15 lines 41-46).

As to claim 22, Johnson discloses the method of reducing noise wherein the entire process is repeated responsive to the presence of additional input speech signal frames (Johnson a state transition diagram of the enclosed invention where the process is repeated upon receiving speech signals, figure 2).

As to claim 43, Johnson discloses the use of ramped noise (ramped noise is defined as a non-stationary noise like a car passing at a distance. Johnson teaches using non-stationary noise that can come from a passing car, column 3 lines 14-19).

As to claim 44, Johnson discloses the use of non-stationary noise (a non-stationary noise like a car passing at a distance; Johnson teaches non-stationary noise that can come from a passing car, column 3 lines 16-19 and column 5 lines 15-22).

3. Claims 20, 21, 23, 24-26, 34-44 are rejected under 35 U.S.C. 103(a) as being unpatentable over Johnson (US 6,415,253) in view of Adlersberg et al. (US 5,012,519) as applied to claims 1, 23, 28 and 35 and in further view Sluijter et al. (US 6,985,855).

As to claim 20, Johnson discloses using the average noise from the input signal (see column 8 lines 46-55). Johnson does not disclose specifically average noise is filtered from the noise reduced output signal frame.

Sluijter teaches a post-process of filtering an output signal that passes through a synthesis filter and using a threshold value to see if noise exists in the signal (column 2 lines 47-59). When noise is detected it is then reduced (column 11 lines 50-60).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the post-process of Sluijter. Making the post-processing means dependent of the background noise the speech quality can be improved (Sluijter, column 2 lines 55-60).

As to claim 21, Johnson does not disclose specifically the step of filtering said average noise comprises adapting a post-processed noise level to an acceptable level.

Sluijter teaches a post-process of filtering an output signal that passes through a synthesis filter and using a threshold value to see if noise exists in the

signal (column 2 lines 47-59). When noise is detected it is then reduced, column 11 lines 50-60.

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the post-process of Sluijter. Making the post-processing means dependent of the background noise the speech quality can be improved (Sluijter, column 2 lines 55-60).

As to claim 23, Johnson discloses a method of filtering a noise component from an input speech signal comprised of signal frames the improvement (abstract lines 1-10) comprising the steps of: estimating said noise component present in the speech signal (noise activity, which is the noise signal is detected from the input channel for the use of signal-to-noise ratio, see column 1 lines 57 and 58); modifying said input speech signal based on an estimation of the noise component (Johnson teaches applying the gain factor to in order to modify the frequency components, the frequency components are received from the windowed frame of signals, column 3 lines 50 and 51, in order to attenuate the spectrum, column 3 lines 55-60); identifying speech segment from said noise component (voice activity, which is the speech signal, is detected from the input channel for the use of signal-to-noise ratio, column 1 lines 56 and 57) as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ratio thresholds (computing the average energy of the

overall signals then compares it to the incoming signals and analyzes the relationship between the energies of which are composed of calculated SNR values and used to determine levels of SNR in the received signal and the average signals, column 8 lines 46-67).

Johnson does not specifically disclose adapting a post-processed noise component to an acceptable, noise-reduced level.

Sluijter teaches a post-process of filtering an output signal that passes through a synthesis filter and using a threshold value to see if noise exists in the signal (see column 2 lines 47-59). When noise is detected it is then reduced, column 11 lines 50-60.

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the post-process of Sluijter. Making the post-processing means dependent of the background noise the speech quality can be improved (Sluijter, column 2 lines 55-60).

Johnson or Sluijter do not disclose specifically detecting voice activity in said channel as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ration thresholds.

Ashley teaches a system that overcomes the problem of fluctuation of non-stationary noise in speech signals (abstract) by using a Voice Activity Detection system for detecting voice in a signal (column 2 lines 31-33) the system then can detect voice activity by tracking the SNR of incoming signals

and determines VAD SNR threshold of which is then biased using average variation in the incoming SNR values. This is done by estimating a SNR of an input signal (column 2 lines 34-36) and a threshold value is generated by the system based on the estimated SNR and biasing the VAD threshold based on a variation of the estimated SNR (column lines 36-38). Next, the threshold values are used to compare to the incoming calculated SNR values (column 8 lines 29-35). Finally, the averaged long-term power spectral estimates are performed (column 6 lines 62-64) in order to calculate the spectral deviation between the power spectrum and the average long-term power spectral estimate (column 6 lines 55 and 56). The deviation estimates are calculated from the total energy estimates that are derived from the channel energy estimate calculations (column 6 lines 25-27) of which are used to calculate the signal to noise ratio from the channel energy (column 5 lines 55-60).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method of Johnson and Adlersberg with the method of detecting voice activity in said channel as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ratio thresholds as taught by Ashley. Using thresholds of average SNR ratios to compare to the incoming SNR from the signal allows for the threshold value to vary according to the specific noise environment that change and a consistent threshold value would not function correctly in an environment such as a car that outputs non-stationary noise. In

this environment the VAD proposed will avoid loss of channel capacity or signal for CDMA variable rate coding as well as loss of signal for fixed-rate TDM with DTX (Ashley, column 1 lines 26-37) and a threshold value that is geared to decrease noise of non-stationary noise (Ashley, abstract) increases the ability to reduce noise in an environment such as in a car.

As to claims 24, and 42, Johnson discloses ramped noise (a non-stationary noise like a car passing at a distance; Johnson teaches non-stationary noise that can come from a passing car, column 3 lines 16-19).

As to claims 25, and 43, Johnson does not specifically disclose said noise component is Gaussian. Adlersberg teaches reducing noise signal from a speech signal; the noise signal being a Gaussian signal (column 7 lines 25-28).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the Gaussian signal in Adlersberg. Gaussian noise signals are a very common noise signal and in order to effectively reduce all noise signals, Gaussian noise signals would also have to be included.

As to claims 26, and 44, Johnson discloses non-stationary noise (see column 3 lines 16 & 17).

As to claim 34, Johnson does not disclose specifically said second filtering means adapts a post-processed noise level to an acceptable level.

Sluijter teaches a post-process of filtering an output signal that passes through a synthesis filter and using a threshold value to see if noise exists in the signal (column 2 lines 47-59). When noise is detected it is then reduced (column 11 lines 50-60). It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the post-process of Sluijter. Making the post-processing means dependent of the background noise the speech quality can be improved (column 2 lines 55-60).

As to claim 35, Johnson discloses a method of noise cancellation in a received speech signal comprised of signal frames (Johnson teaches a noise suppression, of which cancels noise, device, see abstract lines 1-3) comprising the steps of: applying a windowed Fourier transform to said signal frames (Johnson discloses applying fast Fourier transform to the appended data frames, abstract lines 7-9); estimating a noise component present in said signal frames (noise activity, which is the noise signal, is detected from the input channel/components for the use of signal-to-noise ratio, column 1 lines 57 and 58); modifying said signal frames based on a calculated noise estimate (this is done by producing gain multiplicative factors based on the noise spectral estimate and frequency spectrum components, the frequency components are

received from the windowed frame of signals, column 3 lines 50 and 51, column 3 lines 55-57); identifying speech segments from said noise component (voice activity, which is the speech signal, is detected from the input channel for the use of signal-to-noise ratio, column 1 lines 56 and 57) as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ratio thresholds (computing the average energy of the overall signals then compares it to the incoming signals and analyzes the relationship between the energies of which are composed of calculated SNR values and used to determine levels of SNR in the received signal and the average signals, column 8 lines 46-67).

Johnson does not disclose specifically adapting a post-processed noise level to an acceptable level Sluijter teaches a post-process of filtering an output signal that passes through a synthesis filter and using a threshold value to see if noise exists in the signal (see column 2 lines 47-59). When noise is detected it is then reduced, column 11 lines 50-60.

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the post-process of Sluijter. Making the post-processing means dependent of the background noise the speech quality can be improved (see column 2 lines 55-60).

Johnson or Sluijter do not disclose specifically detecting voice activity in said channel as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ration thresholds.

Ashley teaches a system that overcomes the problem of fluctuation of non-stationary noise in speech signals (abstract) by using a Voice Activity Detection system for detecting voice in a signal (column 2 lines 31-33) the system then can detect voice activity by tracking the SNR of incoming signals and determines VAD SNR threshold of which is then biased using average variation in the incoming SNR values. This is done by estimating a SNR of an input signal (column 2 lines 34-36) and a threshold value is generated by the system based on the estimated SNR and biasing the VAD threshold based on a variation of the estimated SNR (column lines 36-38). Next, the threshold values are used to compare to the incoming calculated SNR values (column 8 lines 29-35). Finally, the averaged long-term power spectral estimates are performed (column 6 lines 62-64) in order to calculate the spectral deviation between the power spectrum and the average long-term power spectral estimate (column 6 lines 55 and 56). The deviation estimates are calculated from the total energy estimates that are derived from the channel energy estimate calculations (column 6 lines 25-27) of which are used to calculate the signal to noise ratio from the channel energy (column 5 lines 55-60).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method of Johnson and Adlersberg with the method of detecting voice activity in said channel as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ratio thresholds as taught by Ashley. Using

thresholds of average SNR ratios to compare to the incoming SNR from the signal allows for the threshold value to vary according to the specific noise environment that change and a consistent threshold value would not function correctly in an environment such as a car that outputs non-stationary noise. In this environment the VAD proposed will avoid loss of channel capacity or signal for CDMA variable rate coding as well as loss of signal for fixed-rate TDM with DTX (Ashley, column 1 lines 26-37) and a threshold value that is geared to decrease noise of non-stationary noise (Ashley, abstract) increases the ability to reduce noise in an environment such as in a car.

As to claim 36, Johnson discloses approximating magnitudes of said signal frames (Johnson discloses producing sets of magnitude components of the frequency spectrum for each of the frames, column 7 lines 20-24); computing Signal-to-Noise Ratio magnitudes of said signal frames (Johnson explains that in spectral subtraction a signal-to-noise ratio is calculated by considering the magnitudes of the speech and noise signal, column 1lines 55-60); detecting any noise component on said channel (noise activity, which is the noise signal, is detected from the input channel for the use of signal-to-noise ratio, column 1 lines 57 and 58); detecting stepping noise component on said channel (noise activity, which is the noise signal, is detected from the input channel for the use of signal-to-noise ratio, column 1 lines 57 and 58) and the noise component is a stepping noise like a non-stationary noise (ramped noise is defined as a non-

stationary noise like a car passing at a distance. Johnson teaches using non-stationary noise that can come from a passing car, column 3 lines 14-19); and estimating a gain in said noise component (this is done by producing gain multiplicative factors based on the noise spectral estimate and frequency spectrum components, the frequency components are received from the windowed frame of signals, column 3 lines 50 and 51 and column 3 lines 55-57).

As to claim 37, Johnson discloses noise components comprising ramping noise components, non-stationary noise components, or both (ramped noise is defined as a non-stationary noise like a car passing at a distance. Johnson teaches using non-stationary noise that can come from a passing car, column 3 lines 14-19).

As to claim 38, Johnson does not specifically disclose computing a spectral gain function from an estimated noise history.

Adlersberg teaches a noise reduction system (abstract), of which applies an estimated noise history to signal frames to compute a spectral gain function (the gain function that is calculated depends on the prior and posterior signal-to-noise ratios, which are historical values, column 7 lines 45-50);

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method of Johnson with historic SNR values for gain function calculation as disclosed by Adlersberg.

Doing so would have given an accurate statistical estimator of how the gains in each frame should change according to the past SNR values.

As to claim 39, Johnson discloses applying an inverse Fourier transform thereby reconstructing noise reduced signal frames (which is an inverse Fast Fourier Transform, column 1 lines 62-64, is used to convert the frequency components of the frames into time domain and reconstructing the noise reduced signal, column 3 lines 59-61). Johnson also discloses applying said spectral gain function to the components of a Fourier transform of said signal frames (Johnson teaches applying the gain factor to the frequency components, the frequency components are received from the windowed frame of signals, column 3 lines 50 and 51) in order to attenuate the spectrum, column 3 lines 55-60). Johnson teaches applying said spectral gain function to the real and imaginary components of a Fourier transform of said signal frames.

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method of Johnson with the description of real and imaginary components in its Fourier transform calculation, since Johnson discloses using the frequency components with the Fourier transform and a Wiener filter, the Fourier transform that is used with the Wiener filter includes real and imaginary parts in its calculation in order to describe the speech signal.

As to claim 40, Johnson discloses identifying speech segments from said noise component further comprises applying a windowed Fourier transform on an output signal frame (once the noise has been reduced from the signal an inverse transformer, which is an inverse Fast Fourier Transform, column 1 lines 62-64, is used to convert the frequency components of the frames into time domain and reconstructing the noise reduced signal, column 3 lines 59-61).

As to claim 41, Johnson does not specifically disclose adapting a post-precessed noise component to an acceptable level further comprises filtering average noise from an output signal frame.

Sluijter teaches a post-process of filtering an output signal that passes through a synthesis filter and using a threshold value to see if noise exists in the signal (column 2 lines 47-59). When noise is detected it is then reduced (column 11 lines 50-60).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the post-process of Sluijter. Making the post-processing means dependent of the background noise the speech quality can be improved (column 2 lines 55-60).

As to claim 42, Johnson discloses the use of Gaussian noise component (Johnson refers to a related art that uses similar method to reduce noise and the

noise component is an additive white noise). It would have been obvious to one having ordinary skill in the art at the time the invention was made that a Gaussian noise component that has a probability density function, which is commonly used as additive white noise to yield additive white Gaussian noise.

4. Claim 27 is rejected under 35 U.S.C. 103(a) as being unpatentable over Johnson (US 6,415,253) in view of Adlersberg et al. (US 5,012,519) as applied to claim 1 and in further view of Hermansky et al. (US 6,098,038).

As to claim 27, Johnson discloses applying said spectral gain function to the real and imaginary components of a Fourier transform of said input speech signal (Johnson teaches applying the gain factor to the frequency components, the frequency components are received from the windowed frame of signals, column 3 lines 50 and 51, in order to attenuate the spectrum, column 3 lines 55-60); and, processing said Fourier transform by an inverse Fourier transform thereby reconstructing a noise reduced speech signal (once the noise has been reduced from the signal an inverse transformer, which is an inverse Fast Fourier Transform, column 1 lines 62-64, is used to convert the frequency components of the frames into time domain and reconstructing the noise reduced signal, column 3 lines 59-61).

Johnson or Adlersberg does not specifically disclose using a histogram. Hermansky teaches using an estimated noise histogram and/or a generated noise histogram compute a spectral gain function (this is done by computing the histogram of the noise signal's amplitudes. The peak of the smoothed histogram is chosen as the noise amplitude estimate. These calculations are further used to estimate the noise-to-signal ratio, column 4 lines 35-43).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the spectral gain function method of Johnson with the use of histogram calculations as disclosed by Hermansky. Doing so would allow making a more accurate estimation of a stationary noise signal level and clean speech signal. Since the power of the clean speech signal is unknown, the power of the available noisy signal can be used, thus obtaining an estimate of the noisy signal-to-noise ratio, Hermansky, column 4 lines 39-41).

5. Claim 11 is rejected under 35 U.S.C. 103(a) as being unpatentable over Johnson (US 6,415,253) in view of Adlersberg et al. (US 5,012,519) as applied to claims 1, and in further view of Bizjak et al. (US PGPub 2003/0035549).

As to claim 11, Johnson and Adlersberg do not specifically disclose sampling a slew rate of said noise reduced output signal frame. Bizjak teaches a signal processing method where the slew rate is calculated and applied to non-

linear smoothing which keeps the output signal under the clipping threshold (see paragraph [0326] lines 12-17).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the spectral gain function method of Johnson with the use of the calculation of the slew rate as disclosed by Bizjak, by doing so it would have been easier to detect ramping noise signals since the slew rate calculate the rate of change of the noise signal in time (Bizjak, paragraph [0326] lines 14-18).

6. Claims 28-32 are rejected under 35 U.S.C. 103(a) as being unpatentable over Johnson (US 6,415,253) in view of Adlersberg et al. (US 5,012,519) as applied to claims 1, and in further view of Bizjak et al. (US PGPub 2003/0035549).

As to claim 28, Johnson discloses a system for noise cancellation (abstract lines 1-3) comprising: a first input means operable connected to a processor said first input means receiving a speech signal (Johnson explains that in spectral subtraction an input signal is received and converted to components, column 1 lines 47-50, by a digital signal processor, column 5 lines 60-67); an output means operable connected to said first and second input means and said output speech signal (The noise suppression device modifies magnitude of the time domain data based on the voicing information outputted form the voice

activity detector, abstract 16-19); and, a processing means operably connected to said first and second input means and said output means (this is done by the digital signal processor, column 5 lines 60-67), said processing means comprising a control and storage means (column 6 lines 1-6), a first filtering means, a second filtering means (a first or pre-filter is used to remove dc components and a second, Wiener, filter is used to smoothen the signals (abstract), a voice activity detector (abstract lines 18 and 19), and a sampling and adjustment means (the sampling is done by the breaking of the signals into components (abstract lines 1-5) and the adjusting is done by the automatic gain control module (figure1 # 30), said voice activity detector detects and attacks noise activity on a frequency channel as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ratio thresholds (computing the average energy of the overall signals then compares it to the incoming signals and analyzes the relationship between the energies of which are composed of calculated SNR values and used to determine levels of SNR in the received signal and the average signals, column 8 lines 46-67).

Johnson does not specifically disclose using a second input means operably connected to said processor wherein historical speech and noise data may be entered into a control and storage means for access by said processor. Adlersberg teaches a noise reduction system (abstract), of which applies an estimated noise history to signal frames to compute a spectral gain function (the gain function that is calculated depends on the prior and posterior signal-to-noise

ratios, which are historical values, column 7 lines 45-50); Johnson also does not disclose specifically using a noise step detector. Bizjak teaches a signal processing method where the slew rate is calculated and applied to non-linear smoothing which keeps the output signal under the clipping threshold (see paragraph [0326] lines 12-17).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method of Johnson with historic SNR values for gain function calculation as disclosed by Adlersberg and with the use of the calculation of the slew rate as disclosed by Bizjak. Doing so would have given an accurate statistical estimator of how the gains in each frame should change according to the past SNR values and it would have also been easier to calculate ramping noise signals since the slew rate calculate the rate of change of the noise signal in time (Bizjak, paragraph [0326] lines 14-18).

Johnson or Adlersberg do not disclose specifically wherein asid voice activity detector detects and attacks noise activity on a frequency channel as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ratio thresholds.

Ashley teaches a system that overcomes the problem of fluctuation of non-stationary noise in speech signals (abstract) by using a Voice Activity Detection system for detecting voice in a signal (column 2 lines 31-33) the system then can detect voice activity by tracking the SNR of incoming signals

and determines VAD SNR threshold of which is then biased using average variation in the incoming SNR values. This is done by estimating a SNR of an input signal (column 2 lines 34-36) and a threshold value is generated by the system based on the estimated SNR and biasing the VAD threshold based on a variation of the estimated SNR (column lines 36-38). Next, the threshold values are used to compare to the incoming calculated SNR values (column 8 lines 29-35). Finally, the averaged long-term power spectral estimates are performed (column 6 lines 62-64) in order to calculate the spectral deviation between the power spectrum and the average long-term power spectral estimate (column 6 lines 55 and 56). The deviation estimates are calculated from the total energy estimates that are derived from the channel energy estimate calculations (column 6 lines 25-27) of which are used to calculate the signal to noise ratio from the channel energy (column 5 lines 55-60).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method of Johnson and Adlersberg with the method of detecting voice activity in said channel as a function of conditional comparisons of received signal-to-noise ratios and average signal-to-noise ratio thresholds as taught by Ashley. Using thresholds of average SNR ratios to compare to the incoming SNR from the signal allows for the threshold value to vary according to the specific noise environment that change and a consistent threshold value would not function correctly in an environment such as a car that outputs non-stationary noise. In

this environment the VAD proposed will avoid loss of channel capacity or signal for CDMA variable rate coding as well as loss of signal for fixed-rate TDM with DTX (Ashley, column 1 lines 26-37) and a threshold value that is geared to decrease noise of non-stationary noise (Ashley, abstract) increases the ability to reduce noise in an environment such as in a car.

As to claim 29, Johnson discloses first filtering means filters Signal-to-Noise Ratio magnitudes and signal magnitudes (Johnson clearly explains that in the first process of spectral subtraction for noise suppression a filter is used to estimate the power spectral density, thereby generating a signal-to-noise ratio, column 1 lines 52-55).

As to claim 31, Johnson discloses the noise activity being ramping, non-stationary, or both (ramped noise is defined as a non-stationary noise like a car passing at a distance. Johnson teaches using non-stationary noise that can come from a passing car, column 3 lines 14-19).

As to claim 32, Johnson does not disclose specifically the noise step detector detects and attacks a stepping noise component on said frequency channel.

Bizjak teaches a signal processing method where the slew rate is calculated and applied to non-linear smoothing which keeps the output signal under the clipping threshold (paragraph [0326] lines 12-17).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the spectral gain function method of Johnson with the use of the calculation of the slew rate as disclosed by Bizjak, by doing so it would have been easier to calculate ramping noise signals since the slew rate calculate the rate of change of the noise signal in time (Bizjak, paragraph [0326] lines 14-18).

7. Claims 16 and 17 are rejected under 35 U.S.C. 103(a) as being unpatentable over Johnson (US 6,415,253) in view of Adlersberg et al. (US 5,012,519) as applied to claims 1, 23, 28 and 35 and in further view Sluijter et al. (US 6,985,855) and in further view of Bizjak et al. (US PGPub 2003/0035549) and Hermansky et al. (US 6,098,038).

As to claim 16, Johnson discloses using the average noise from the input signal (see column 8 lines 46-55).

Johnson does not disclose specifically average noise is filtered from the noise reduced output signal frame.

Sluijter teaches a post-process of filtering an output signal that passes through a synthesis filter and using a threshold value to see if noise exists in the signal (column 2 lines 47-59). When noise is detected it is then reduced (column 11 lines 50-60).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the post-process of Sluijter. Making the post-processing means dependent of the background noise the speech quality can be improved (see column 2 lines 55-60).

As to claim 17, Johnson does not disclose specifically the step of filtering said average noise comprises adapting a post-processed noise level to an acceptable level.

Sluijter teaches a post-process of filtering an output signal that passes through a synthesis filter and using a threshold value to see if noise exists in the signal (column 2 lines 47-59). When noise is detected it is then reduced (column 11 lines 50-60).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise signal disclosed in Johnson with the post-process of Sluijter. Making the post-processing means dependent of the background noise the speech quality can be improved (column 2 lines 55-60).

8. Claims 12-15, 18, and 33 are rejected under 35 U.S.C. 103(a) as being unpatentable over Johnson (US 6,415,253) in view of Adlersberg et al. (US 5,012,519)

as applied to claims 1, and in further view of Bizjak et al. (US PGPub 2003/0035549) and Hermansky et al. (US 6,098,038).

As to claim 12, Johnson discloses deciding to continue said sampling, the sampling can be done a 8kHz and will continue until a desired amount of bits have been sampled. The system decides to continue sampling until the desired sampling bits are accomplished (column 6 lines 13-20).

Johnson does not specifically disclose using a counter, slew rate, or histogram.

Bizjak teaches using a clip event counter as an input and updating the counter by using a reset clip counter, in order to reset a counter the counter would have to be started initially (table A in paragraph [0237]). Bizjak also teaches adjusting the sample slew rate; the slew rate is calculated and applied to non-linear smoothing which keeps the output signal under the clipping threshold (paragraph [0326] lines 12-17).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method as disclosed by Johnson with the counter and slew rate of Bizjak. Doing so would allow keeping track of the slew rate with the counter allowing detecting non-stationary noise.

Johnson or Bizjak do not disclose specifically using a histogram.

Hermansky teaches encoding and a noise sample and decoding a noise estimate, in order for the noise signal to be used for further calculations it would have to be encoded from an input signal (column 4 lines 35-37) and decoded in its raw form for further analysis. Hermansky also teaches using an estimated noise histogram and/or a generated noise histogram compute a spectral gain function (this is done by computing the histogram of the noise signal's amplitudes. The peak of the smoothed histogram is chosen as the noise amplitude estimate. These calculations are further used to estimate the noise-to-signal ratio, column 4 lines 35-43), every time that a histogram calculation would be used it would have to be updated so it can represent the new data and normalized to the new setting so the calculations can be accurate. A weighted histogram bin is represented by a number of amplitudes that have been represented by histogram calculations and have certain weights or importance and the peak amplitude is chosen as the noise amplitude.

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the noise reduction method as disclosed by Johnson with the histogram of Hermansky. Doing so would allow keeping track of the slew rate with the counter allowing detecting non-stationary noise and using a histogram would allow making a more accurate estimation of a stationary noise signal level and clean speech signal. Since the power of the clean speech signal is unknown, the power of the available noisy signal can be

used, thus obtaining an estimate of the noisy signal-to-noise ratio, Hermansky column 4 lines 39-41).

As to claim 13, Johnson and Adlersberg do not specifically disclose measuring an error period.

Bizjak teaches sampling a slew rate to make a distinction of speed of smoothing (paragraph [0327] lines 1-10) and calculating the error of speed tracking (paragraph [0338] lines 20-24).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the spectral gain function method of Johnson with the use of the calculation of the error as disclosed by Bizjak, by doing so it would have been easier to calculate ramping noise signals with the use of errors and threshold values (paragraph [0338] lines 20-24).

As to claim 14, Johnson and Adlersberg do not specifically disclose a counter reset.

Bizjak teaches using a clip event counter as an input and a reset clip counter as outputs (table A in paragraph [0237]).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the spectral gain function method of Johnson with the use of a counter reset as disclosed by Bizjak. Doing so

would allow calculating new values that do not carry old values that might compromise the integrity of the present values.

As to claim 15, Johnson discloses noise reduced output signal frame is overlapped and added to previous noise reduced output signals frame (Johnson teaches overlapping adjacent frames (column 15 lines 33-36) and using the immediately previous frame (column 15 lines 41-46).

As to claim 18, Johnson discloses the method of reducing noise wherein the entire process is repeated responsive to the presence of additional input speech signal frames (Johnson a state transition diagram of the enclosed invention where the process is repeated upon receiving speech signals, figure 2).

As to claim 33, Johnson and Adlersberg do not disclose specifically sampling and adjusting means samples and adjusts a slew rate and a histogram of said output speech signal.

Bizjak teaches a signal processing method where the slew rate is calculated and applied to non-linear smoothing which keeps the output signal under the clipping threshold (paragraph [0326] lines 12-17).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the spectral gain function method of Johnson with the calculation of the slew rate as disclosed by Bizjak. Doing so

would have allowed making an accurate estimation of the noise signal and cleaning speech signal. By the used of the slew rates it would have been easier to calculate ramping noise signals (Bizjak paragraph [0326] lines 14-18).

Johnson, Adlersberg, or Bizjak do not teach using histograms.

Hermansky teaches using an estimated noise histogram and/or a generated noise histogram compute a spectral gain function (this is done by computing the histogram of the noise signal’s amplitudes. The peak of the smoothed histogram is chosen as the noise amplitude estimate. These calculations are further used to estimate the noise-to-signal ratio) (see column 4 lines 35-43).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to have modified the spectral gain function method of Johnson with the use of histogram calculations as disclosed by Hermansky. Doing so would have allowed making an accurate estimation of the noise signal and cleaning speech signal. Since the power of the clean speech signal is unknown, the power of the available noisy signal can be used, thus obtaining an estimate of the noisy signal-to-noise ratio (Hermansky column 4 lines 39-41).

Conclusion

Any inquiry concerning this communication should be directed to Josiah Hernandez whose telephone number is 571-270-1646. The examiner can normally be reached from 7:30 pm to 5:00 pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Xiao Wu can be reached on (571) 272-7761. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

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JH

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6/19/2008